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(54) Multi-phonic balancer

(57) Apparatus maintains the subjective balance for a listener 600 amongst the channels of a multi-phonic e.g. stereophonic, sound reproduction facility as the listener moves around a space served by the facility. Measuring devices 400, 500 for respective speakers 100, 200 are arranged to determine the relative distances of the speakers from the listener by means of signals reflected from the listener, and control means 300 are arranged to vary the respective volumes of sound reproduction from said channels according to said relative distances. The distance measurement can use ultrasonic or infra red techniques.

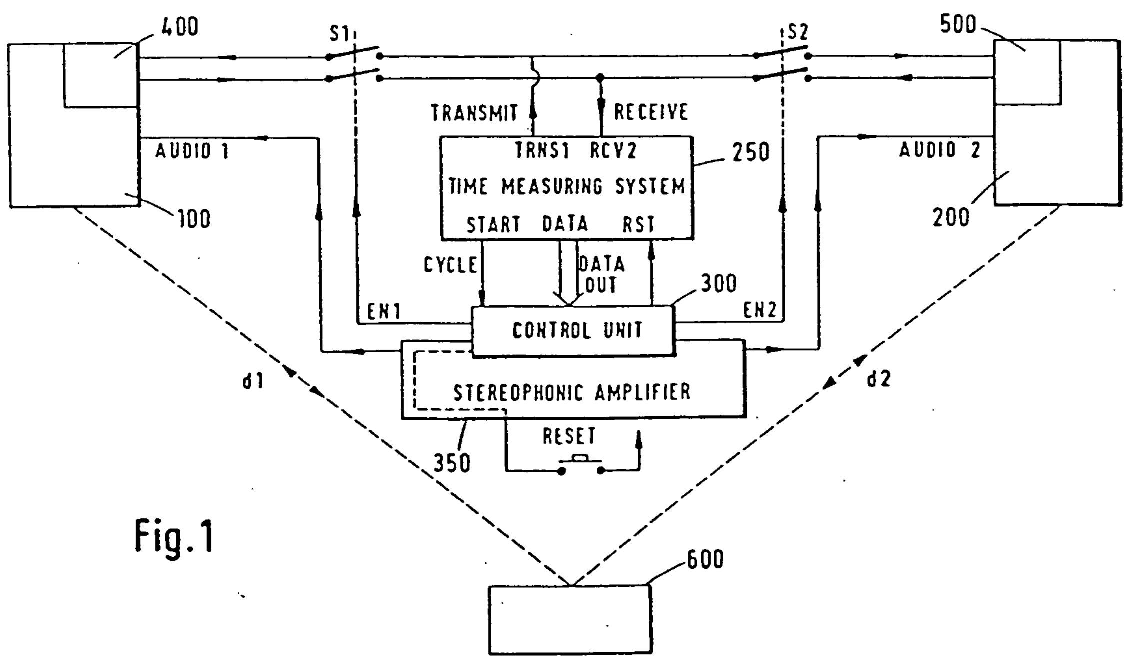


Fig.1

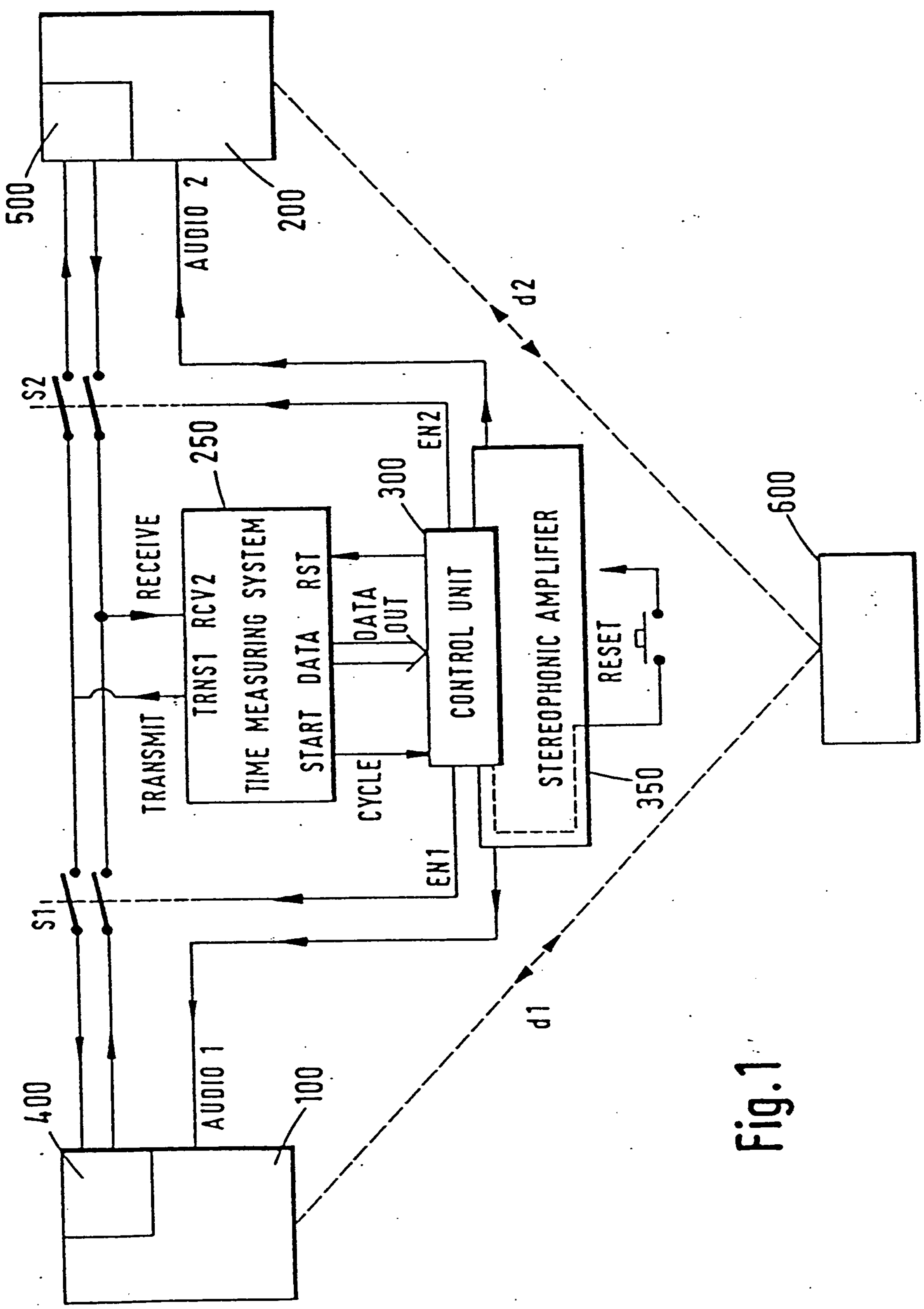

500
200
AUDIO 2
d2
S2
250
RECEIVE
TRNS1 RCV2
TIME MEASURING SYSTEM
START DATA RST
DATA OUT
300
EN2
CONTROL UNIT
STEREOPHONIC AMPLIFIER
RESET
600
TRANSMIT
CYCLE
350
EN1
S1
d1
AUDIO 1
400
100

Fig.1

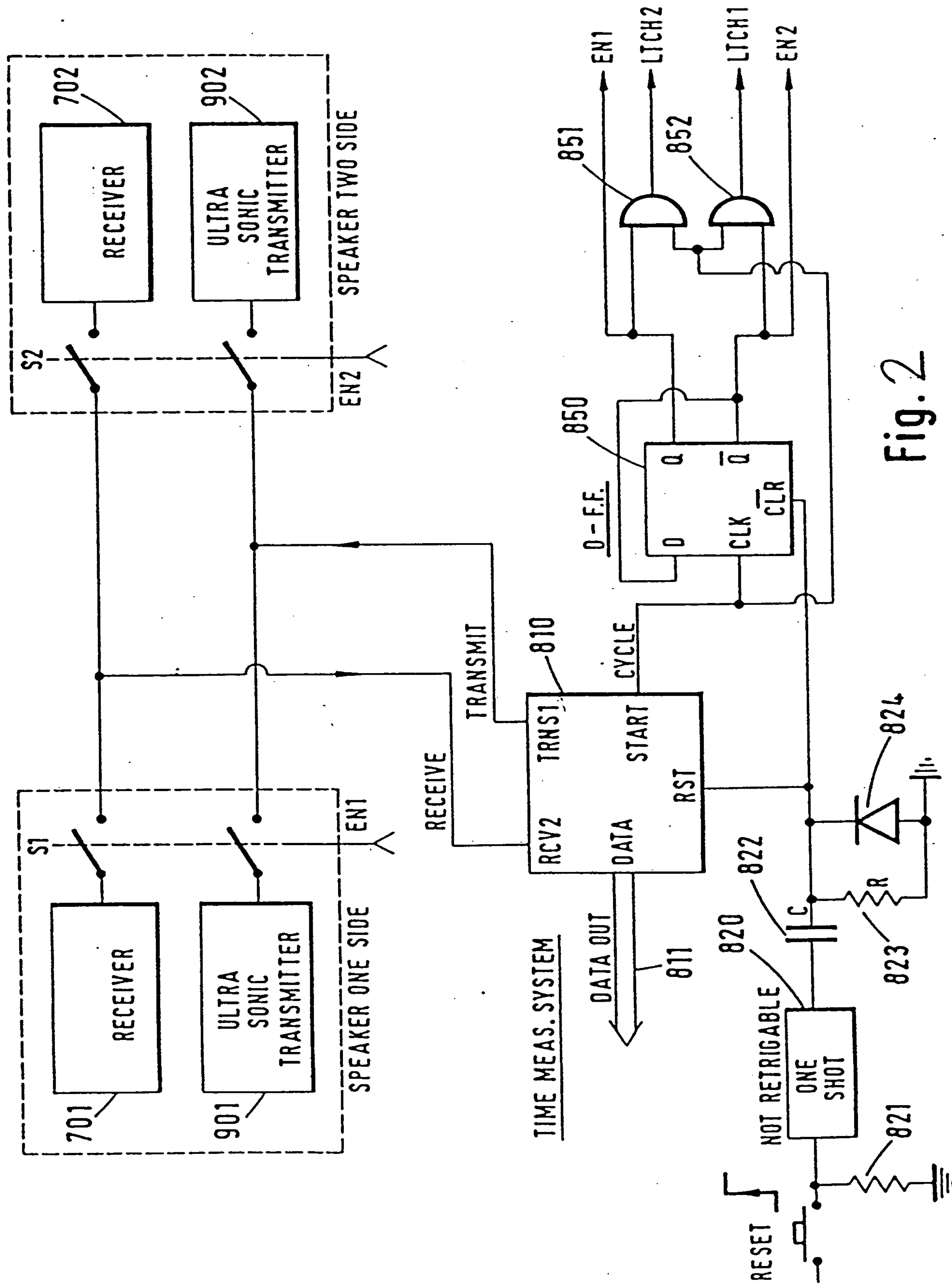

702
RECEIVER
902
ULTRA
SONIC
TRANSMITTER
SPEAKER TWO SIDE
S2
EN2
S1
EN1
RECEIVER
ULTRA
SONIC
TRANSMITTER
SPEAKER ONE SIDE
701
901
RECEIVE
TRANSMIT
810
TRNS1
RCV2
DATA
START
RST
CYCLE
TIME MEAS. SYSTEM
DATA OUT
811
D-F.F.
850
D
Q
CLK
CLR
Q
851
852
EN1
LTCH2
LTCH1
EN2
824
822
C
R
823
820
NOT RETRIGABLE
ONE
SHOT
821
RESET

Fig. 2

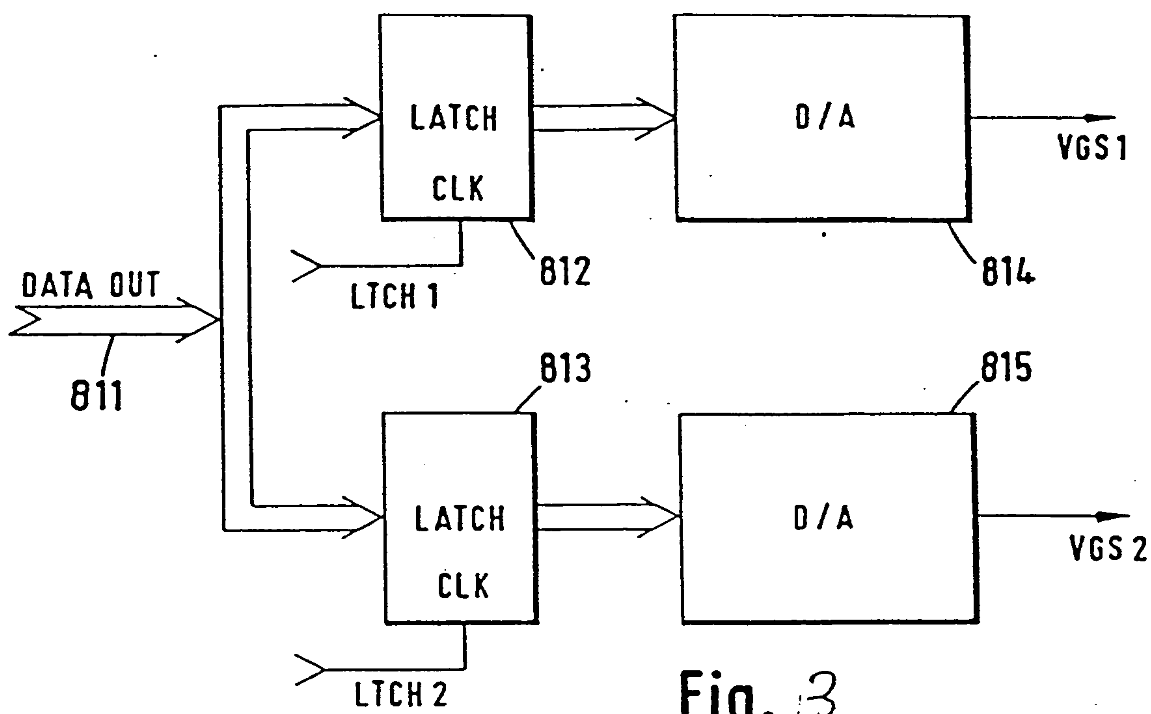

LATCH
CLK
D/A
VGS 1
812
814
DATA OUT
LTCH 1
811
813
815
LATCH
CLK
D/A
VGS 2
LTCH 2

Fig. 3

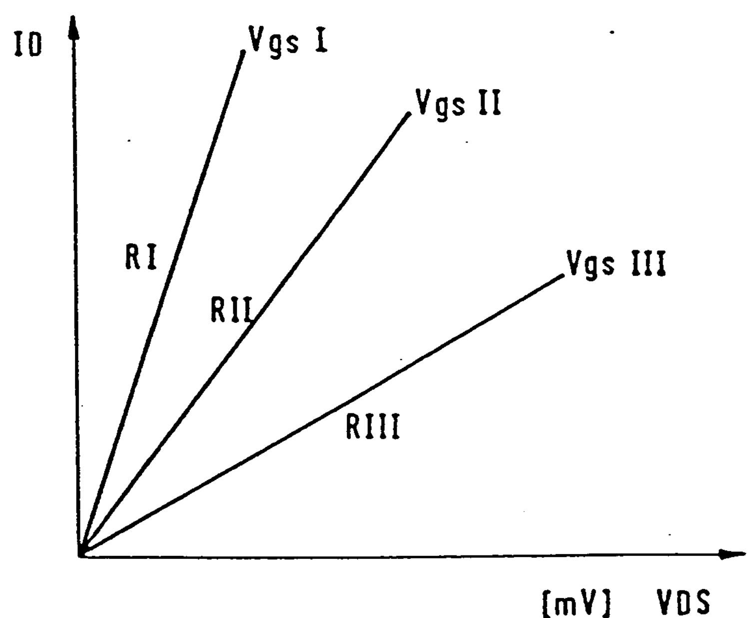
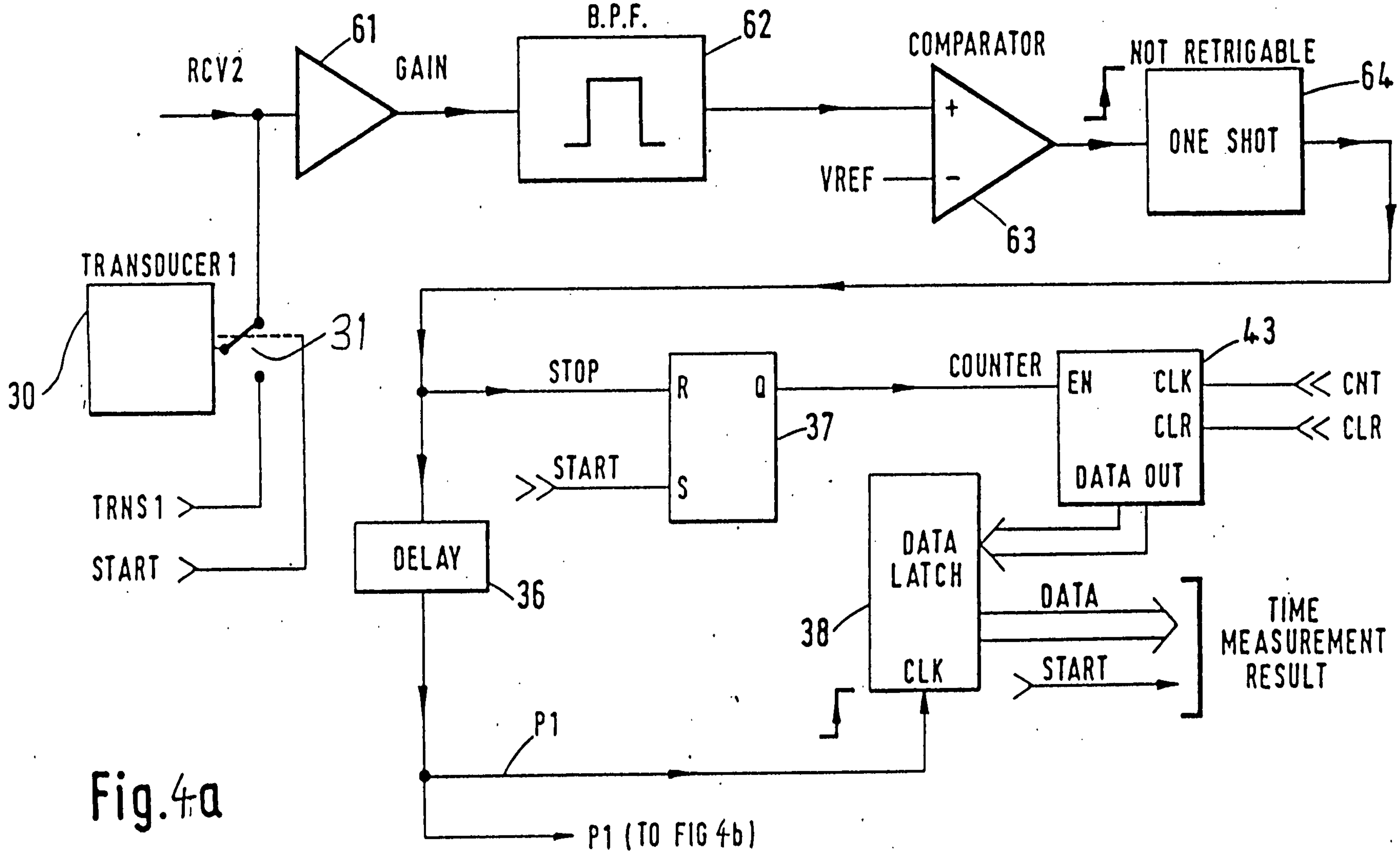

ID
Vgs I
Vgs II
Vgs III
RI
RII
RIII
[mV] VDS

Fig. 5a



RCV2
61
GAIN
B.P.F.
62
COMPARATOR
+
-
VREF
63
NOT RETRIGABLE
ONE SHOT
64
TRANSDUCER 1
30
31
TRNS 1
START
STOP
R
Q
S
START
37
COUNTER
EN
CLK
CLR
DATA OUT
43
CNT
CLR
DELAY
36
DATA LATCH
CLK
38
DATA
START
TIME MEASUREMENT RESULT
P1
P1 (TO FIG 4b)

Fig.4a

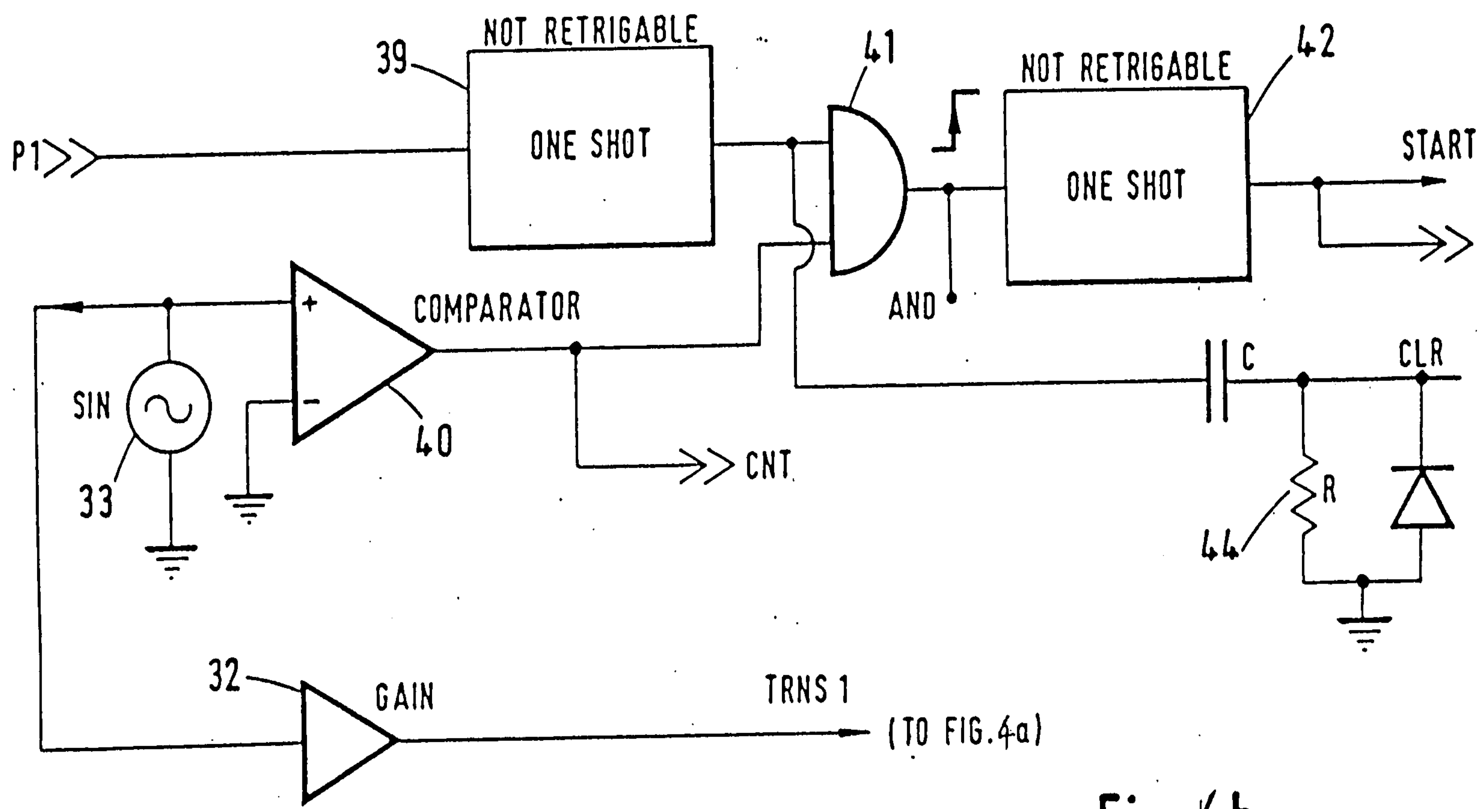
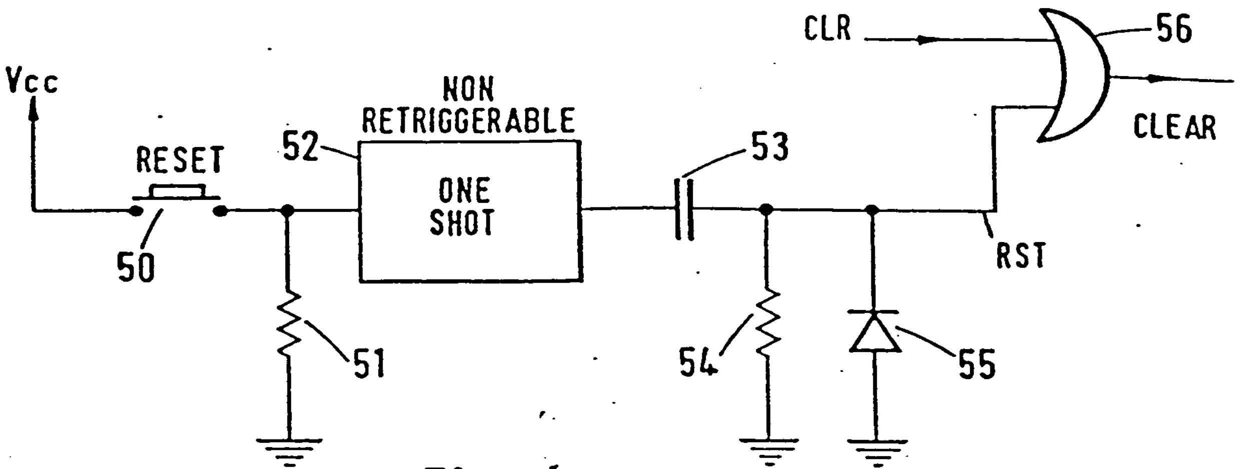

NOT RETRIGABLE
39
ONE SHOT
P1
41
NOT RETRIGABLE
42
ONE SHOT
START
AND
COMPARATOR
SIN
33
40
CNT
C
CLR
R
44
32
GAIN
TRNS 1
(TO FIG.4a)

Fig.4b



CLR
56
Vcc
NON RETRIGGERABLE
CLEAR
53
52
RESET
ONE SHOT
50
RST
51
54
55

Fig. 4c

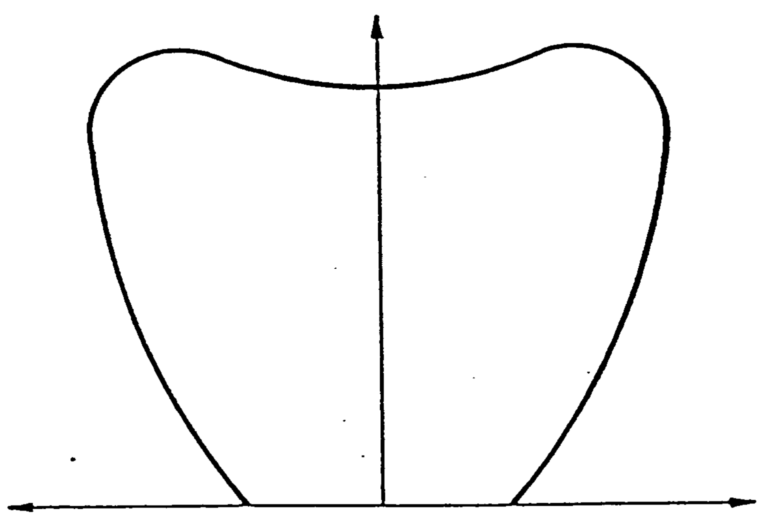


Fig. 8

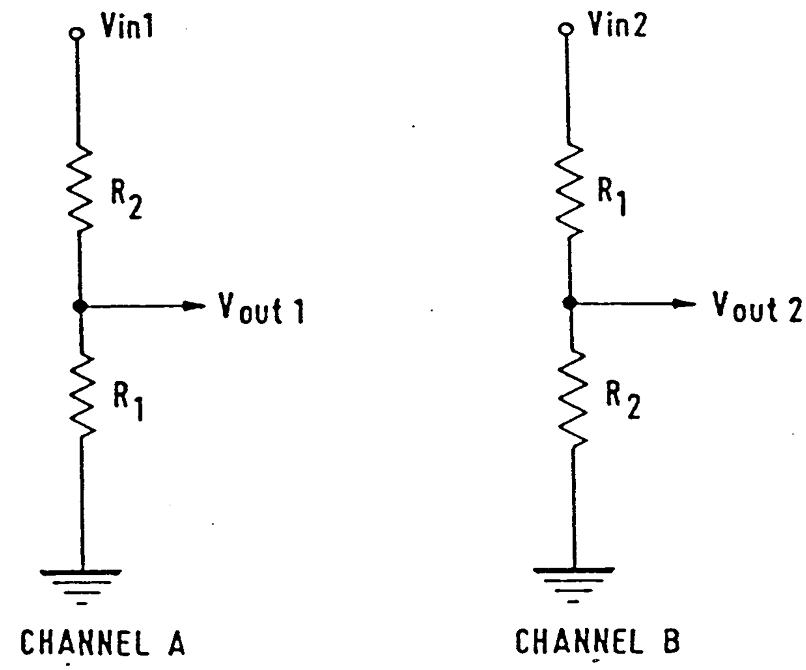

Vin1
R2
Vout 1
R1
CHANNEL A
Vin2
R1
Vout 2
R2
CHANNEL B

Fig. 6

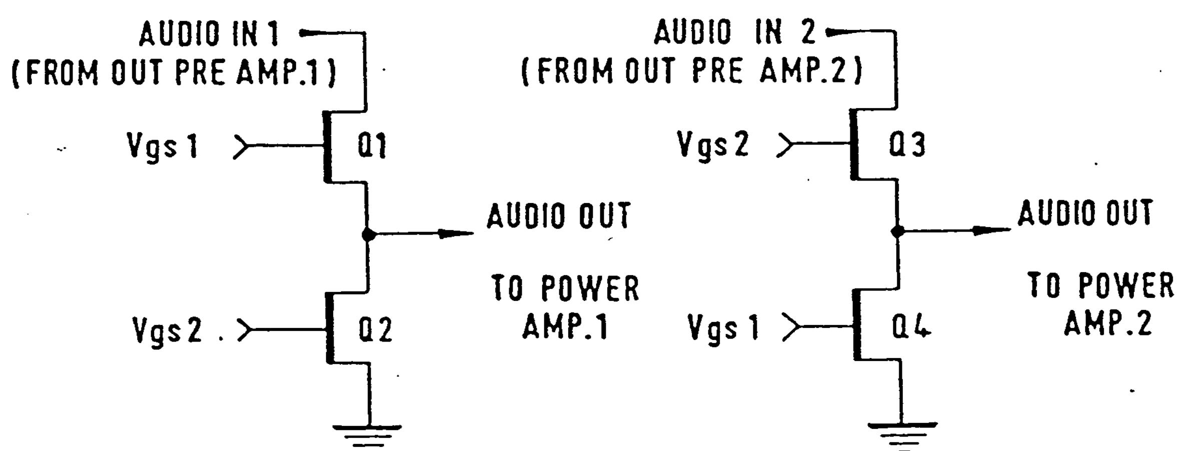

AUDIO IN 1
(FROM OUT PRE AMP.1)
Vgs 1
Q1
AUDIO OUT
TO POWER
AMP.1
Vgs 2
Q2
AUDIO IN 2
(FROM OUT PRE AMP.2)
Vgs 2
Q3
AUDIO OUT
TO POWER
AMP.2
Vgs 1
Q4

Fig. 7

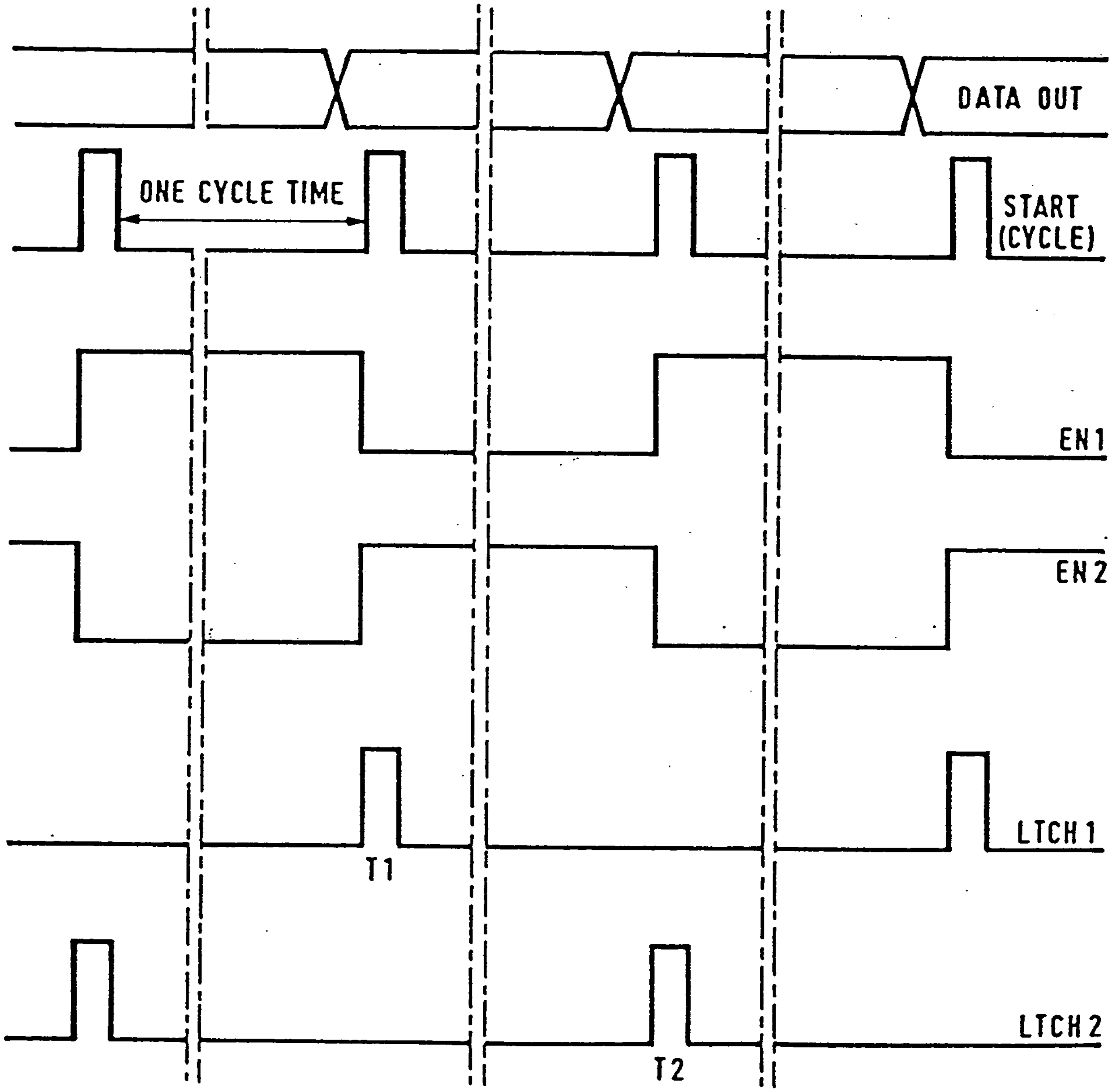


Fig. 5b

MULTI-PHONIC BALANCER

The present invention relates to multi-phonic balancer apparatus and methods and to apparatus and methods for adjusting the subjective balance, for a listener, between the channels of a multi-phonic (e.g. stereophonic) sound reproduction facility to maintain the desired audio effect as the listener moves about the room or other space served by the facility.

This specification describes a modification of a multi-phonic balancer described in the applicant's PCT Application No. US87/00323 filed February 11th, 1987.

At various points in this disclosure, reference is made to ultrasonic waves, to sound, and to pressure waves. For the avoidance of doubt, it is hereby declared that pressure waves of any frequency may be employed within the scope of the present invention, the exact choice depending on the particular application and the circumstances of each individual case, as will be appreciated by those skilled in the art.

It is well known that the spatial illusion created by stereophonic sound reproduction is, for a given balance setting between the channels, effective only over a relatively small zone within the space served

by the system. For example, where the volume settings of two channels of a stereophonic system are equal, the stereophonic effect is normally obtained only at points equi-distant from the speakers.

Consequently, if the listener is moving about the room attending to other matters whilst listening to sound reproduced by the system, the stereo effect will be intermittently lost. One object of the present invention is to provide means whereby the above mentioned defects may be avoided.

According to one aspect of the invention, there is provided apparatus for maintaining the subjective balance for a listener amongst the channels of a multi-phonic (e.g. stereophonic) sound reproduction facility as the listener moves around a space served by the facility, comprising: measuring devices for respective speakers arranged to determine the relative distances of the speakers from the listener by means of signals reflected from the listener; and control means arranged to vary the respective volumes of sound reproduction from said channels according to said relative distances.

In one embodiment, the measuring devices are arranged to transmit at mutually differing frequencies. This permits the individual devices to transmit simultaneously without mutual interference.

In an alternative embodiment, the measuring devices are arranged to transmit at mutually differing times. This embodiment enables the individual devices to utilize the same frequency without mutual interference.

Thus, time division multiplex or frequency division multiplex systems may be employed.

In order to obtain positional data in respect of the unknown location from said time intervals, computing means may be coupled to said timing means.

The spaced reference locations may be fixed relative to the loudspeakers, and said computing means may be arranged to further compute a control signal based on said positional data, control means being provided for controlling the power supply to said speakers in response to said control signal in a manner such as to equalise the audio intensity from respective loudspeakers at said unknown location.

Expediently, the reference locations are within respective housings of said loudspeakers.

According to a further aspect of the invention, there is provided a method for maintaining the subjective balance for a listener amongst the channels of a multi-phonic (e.g. stereophonic) sound reproduction facility as the listener moves around a space served by the facility, in which method measuring devices for respective speakers determine the relative distance of the speakers from the listener by means of signals reflected from the listener, and the respective volumes of sound reproduction from said channels are varied accordingly.

While the invention has been described by reference to a stereophonic system, it will be appreciated that

the balance adjusting system described may be used in conjunction with a system utilizing more than two channels, for example in relation to a quadraphonic system which utilizes four separate channels for sound reproduction.

For a better understanding of the invention, and to show how the same may be carried into effect, reference will now be made by way of example to the accompanying drawings, in which:

Figure 1 illustrates apparatus for stereo balancing according to one embodiment of the invention;

Figure 2 illustrates a block circuit diagram of a circuit for use with the apparatus of Figure 1;

Figure 3 is a block circuit diagram of a further part of the circuit of Figure 2;

Figures 4a and 4b illustrate a block circuit diagram of measuring apparatus for use with embodiments of the present invention;

Figure 4c shows a reset circuit for use with the embodiment of Figures 4a and 4b;

Figure 5a illustrates characteristic curves of a typical JFET;

Figure 5b illustrates waveform diagrams illustrating operation of the circuit of Figures 2 and 3;

Figure 6 is a resistor arrangement for use with the circuit of Figures 2 and 3;

Figure 7 is a circuit arrangement for use with the circuit of Figures 2 and 3 in place of the circuit of Figure 6; and

Figure 8 illustrates a radiation pattern of a loudspeaker.

Referring to Figure 1, there is illustrated schematically apparatus for adjusting automatically the stereo balance perceived by a listener 600 as he moves around a space served by a stereo music reproduction system. The apparatus includes first and second loudspeakers 100 and 200 coupled to a central control unit 300 of a stereophonic reproduction system having an amplifier 350. Associated with the respective loudspeakers are measuring devices 400 and 500. The devices 400 and 500 may be located within the respective speaker housings, or may be mounted in separate units attached to respective speakers.

In addition to the control circuitry 300, there is provided processing circuitry 250 for deriving distance information from the outputs of respective measuring devices 400 and 500 to enable the control unit 300 to control the power supplied by the stereo amplifier 350 to respective speakers 100 and 200.

In outline, the measuring and adjusting process takes place in three stages. First of all, the relative or absolute distances of the listener 600 from the stationary devices 400 and 500 is measured, secondly the effect of these distances on the stereo balance is computed, and thirdly the balance is adjusted in accordance with the computed result.

The devices 400 and 500 preferably each contain a respective sonic range finder, for example as described in US Patent No. 4,192,587 at column 4, line 63. Such range finders emit an ultrasonic pulse which strikes an intended target and generates an echo which may be received by the range finder. The range finder may calculate the distance to the target by measuring the travel time of the pulse to and from the target and by deriving distance based upon knowledge of the speed of the sonic pulse.

The ultrasonic frequencies used are preferably in the range of 50-100 kHz, although the frequency used will depend on the circumstances. To avoid environmental problems, it is preferable that the frequency is above the audible range of any pets likely to be in the vicinity, such as dogs. The system works most effectively when the walls of the room in which it is contained are reasonably non-reflecting. Most domestic living rooms containing curtains, carpets etc. will satisfactorily meet this requirement. This aspect will be discussed in more detail hereinafter.

It is preferred that the transmitters associated with units 400 and 500 have a directional characteristic such that they transmit over a range of angle less than 180 degrees.

Referring now to Figure 2, the construction of one possible embodiment of the control and processing circuitry of Figure 1 will be described in more detail. Blocks 701 and 702 represent ultrasonic receivers, block 810 represents a timing control unit, and blocks 901 and 902 represents ultrasonic transmitters. The system operates to send ultrasonic

interrogation signals to the listener by means of the transducers 901 and 902. When an interrogation signal is transmitted, a timer of block 810 is started. When the reflection pulse is received by receiver 701 or 702, the timer is stopped, and the timed interval is employed as a measure of the distance to the listener from the respective measuring device and the associated loudspeaker.

In addition to the components already described, the apparatus of Figure 2 includes timing logic to ensure correct system sequencing and timing. The object of such timing is to cause signals to be transmitted alternately from the two transmitters 901 and 902 and to ensure that a response signal is received and processed for each transmission before a further transmission is sent. Such timing is provided by a D-type flip-flop 850 in conjunction with two AND gates 851 and 852. As illustrated, the flip-flop 850 is clocked by the signal START from the processing circuit 810 and provides complementary signals EN1 and EN2 on its Q and $\overline{Q}$ outputs. The AND gates receive respective signals EN1 and EN2 and are also connected to receive the signal START. Output signals from the AND gates are designated LTCH1 and LTCH2 and are employed for latching data from a counter of circuit 810 as will be explained with reference to Figure 3. Signals EN1 and EN2 are used to control switches S1 and S2 for enabling the receiver/transmitter pair 701,901 or 702,902. The system timing appears more clearly from Figure 5b which is a waveform diagram illustrating the signals START (CYCLE), EN1, EN2, LTCH1, and LTCH2.

Referring now to Figure 3, two data latches 812 and 813 receive data from the DATA output of the

processing circuit 810 on a data bus 811 in accordance with latching signals LTCH1 and LTCH2. This data is then passed through respective digital-to-analogue converters 814 and 815 to provide signals V_{gs1} and V_{gs2} respectively.

Reference will now be made to Figures 4a and 4b which show a more detailed circuit diagram of apparatus for use in device 400 and 500 of Figure 1 and the associated control and processing circuitry 810 of Figure 2. In Figures 4a and 4b, a single transducer 30 functions as both transmitter 901 and receiver 70. To permit this, a change-over switch 31 is provided.

Upon receipt of a cyclic signal START of pulse length DT, switch 31 operates to allow a sinusoidal signal F_1 to pass from a signal generator 33 via an amplifier 32 to the ultrasonic transducer 30 as signal TRNS1. The resulting ultrasonic wave is reflected from the listener. This echo signal is detected by the transducer 30.

The ultrasound transducer device 30 is then switched to an amplifier 61 which in turn is connected to the input of a band-pass filter 62. The output of the filter 62 is connected to the non-inverting input of a comparator 63 whose inverting input is connected to a reference voltage V_{REF}. A non-retriggerable monostable circuit 64 having a period T_2 receives its input from the comparator 63 and supplies its output to both a delay circuit 36 and to the RESET input of a bistable flip-flop 37 whose SET input is connected to receive signal START. The output of the delay circuit 36 is connected to the clock input of a data latch 38 and to the input of a non-retriggerable monostable multivibrator 39 (Figure 4b).

The circuit operates as follows. After filtering by the band-pass filter 62, the received signal is compared with V_{REF} by comparator 63 which thus sets the receiver sensitivity. The first rising edge of the received signal at the output of the comparator 63 causes the monostable circuit 64 to be triggered. Further triggering of the circuit 64 is not possible until its time period has expired.

It will be appreciated that the pass band of the filter 62 should correspond to the bandwidth DF of the transmitted ultrasound having center frequency F_1 as discussed below.

The detection time of the receiver depends upon V_{REF} at the comparator and may be taken into account when computing the transit time of the ultrasound wave.

The output of the signal generator 33 is also connected to the non-inverting input of a comparator 40 whose inverting input is grounded. The output of the comparator 40 and of the multivibrator 39 are connected to respective inputs of an AND gate 41 whose output AND is connected to the input of a further non-retriggerable monostable multivibrator 42, the output of which provides the signal START.

The Q output of the bistable flip flop 37 is connected to the count input of a counter 43, which is clocked at a frequency F_2 and may be cleared by a signal CLR. Data from the counter 43 is read by the data latch 38 in response to the output signal from the delay circuit 36 and is subsequently supplied as output data representing a timed measurement result. The CLR signal is derived from the output of

monostable circuit 39 via a differentiator 44, across which a diode is connected.

When the ultrasonic wave is received by the transducer 30, the flip flop 37 is reset so that its output Q stops the counting process of the counter 43. After the delay introduced by circuit 36, the data latch 38 reads the counter output. The delay is necessary to ensure safe latching of valid data at the counter output. The delay is slightly longer than the time necessary from generation of signal STOP until the counter output is valid. However, the delay is a very short interval and will be negligible in comparison with the time necessary for sound waves to cover a practical measuring distance. The output of the AND gate 41 will rise only when the following two conditions are fulfilled: (1) The counter output has already been latched and (2) the output of comparator 40 is high. The comparator output will rise only when the sinusoidal output of the clock generator 33 is rising through zero. This ensures that the output of gate 41 will be true only after the data has been latched and when the sinusoidal output of the generator is greater than zero and rising. The interval timed by monostable circuit 39 should be about 2.5 times the period of the generator 33. The output START of monostable circuit 42 will be high for the transmitting period DT and will be synchronised with the transmitted ultrasound wave. Before the signal START goes high, the counter 43 will be cleared by signal CLR and only after it is cleared will the signal START set the R.S. flip-flop 37 to initiate counting. At the same time, the transmitting section is enabled. The generator 33 supplies its output to the transmitter 30 when the switch 51 is closed by the signal START. It will be

understood that the signal START is also available for peripheral circuits wishing to read the data latched in the data latch 38. That is to say, a peripheral circuit may read out the data in response to the START signal. Naturally, in any particular measuring cycle data from the <u>preceding</u> cycle will be read out.

There are a few considerations that should be borne in mind in respect of the frequency control of the ultrasound transmitter. As has been explained, switch 31 is operated by signal START for a controlled interval DT to cause a "pulse" of ultrasound to be transmitted. The length of the transmission interval DT is set by monostable circuit 42. It can be shown by Fourier transform techniques that where a sinewave of frequency F1 is transmitted for an interval DT the resulting transmitted pulse contains a band of frequencies centered on frequency F_1. The majority of the frequencies are however concentrated within a bandwidth DF, where DF=1/DT. It is clear from this relationship that the bandwidth of the pulse is larger as the pulse interval is reduced. The consequence is that the interval DT should be as large as possible. On the other hand, DT cannot be allowed to exceed the the time taken for the ultrasound wave to traverse the distance required to be measured.

In this connection, it is also necessary to consider the effect of DT on the actual frequency transmitted. If it is desired to maintain the transmitted frequency within X% of the center frequency, the minimum transmitted frequency will be 100X/DT. This must be considered when there is a limit on the frequency which may be transmitted.

Generally, the frequency should be maintained as low as possible, because the transmission loss increases as a function of frequency, as will be discussed in more detail hereinafter.

The frequency F_2 of the counter clock depends on the required accuracy. If it is desired to measure distance to within an error of E, then $F_2=V/E$, where V is the velocity of the ultrasound wave.

Figure 4c shows a circuit for use in conjunction with the circuit of Figures 4a and 4b to effect system reset. The output of an OR gate 56 provides a signal for clearing and resetting the counter 43 of Figure 4a. The OR gate 56 has two inputs taken respectively from the output of the debouncing circuit 44 and the output of a similar circuit 55 comprising a capacitor 53, a resistor 54 and a diode 55. The signal input to the capacitor 53 is derived from a monostable circuit 52 which is triggered by operation of a RESET switch 50. The input of the monostable circuit 52 is connected to ground via a resistor 51.

In the illustrated example, the minimum error will be of the order of one wavelength. To improve the accuracy to half a wavelength, a rectifier circuit could be inserted between the amplifier 61 and the filter 62 of Figure 4a. This would permit the comparator 63 to respond at each zero transition of the signal and not only on a positively travelling transition.

The frequency of the counter clock should be selected according to the possible accuracy. If measurements are being made to the nearest wavelength and the center frequency of the return wave is F_2, the

counter clock frequency should be at least F_2. On the other hand, if measurements are made to the nearest half wavelength, the clock frequency should be at least $2F_2$.

In the present embodiment, measures must be taken to ensure that reflections of the first transducer are not interpreted by the associated receiver as transmissions from the other transducer. This problem can be avoided by using different frequency bands for the two transmissions.

Of course, to maintain the stereo balance it is normally only necessary to ensure that substantially equal sound intensities are received by the listener from respective speakers 100 and 200. This will ensure that the stereo effect is maintained irrespective of the position of the listener 600 and irrespective of the <u>absolute</u> values of the sound intensities present. For adjusting the stereo balance, it is therefore only necessary to determine the ratio of the respective distances from the listener to the speakers, and for this purpose the ratio of the two time intervals determined by the timer will suffice.

In some circumstances, a listener may wish to ensure that the intensity from one channel is always greater than that from another. This might apply if one of his ears were less sensitive than the other. Naturally, the system could also be set, within the scope of the invention, to achieve unequal channel balance to meet such a situation.

It remains to be described how the adjustment of the stereo balance is effected in response to the output

signals V_{gs1} and V_{gs2}. This may be achieved either by direct electronic control using, for example a variable resistor such as a JFET connected in the loudspeaker volume control circuitry, or alternatively by using a mechanical system which mechanically rotates a balance control spindle whose rotation effects a necessary change in the stereo balance. Electronic control will be discussed in more detail with reference to Figures 5a, 6 and 7.

In order that the balance between the channels will be continually maintained, the power (in Watts) transmitted from each speaker will have to be changed when the distances d_1 and d_2 are changed. Since the power transmitted from the respective loudspeakers is proportional to the square of the voltages (V_1, V_2) applied across respective loudspeaker impedances, and since the power received by a listener falls off as the square of the distance, it may readily be shown that:

$$V_1/V_2 = d_1/d_2,$$

where d_1 and d_2 are distances from the listener to respective speakers.

Since the final power stage of an audio amplifier (at the input of which stage the balance is generally controlled) has linear behaviour (in the active region), which means that the ratio of the power on each speaker to the power on the final stage input is constant, changes of voltage ratio at the final stage input would have the same effect as changes of voltage ratio at the speakers.

Thus, where U_1 is the input voltage to the first input power amplifier and U_2 is the input voltage to the second input power amplifier, the result is obtained that:

$$U_2/U_1 = d_2/d_1.$$

Therefore, the balance can be controlled by automatically adjusting the ratio U_2/U_1 according to changes in the ratio d_2/d_1.

In a stereophonic system, for each location of the listener in the room, there will be a different value for the ratio U_2/U_1 so that the listener will not need to adjust the balance at all, not even initially. All adjustments may be performed automatically as a function of the listener location at any given moment.

One way to implement the voltages ratio changes would be with JFETS: (active resistances). A set of characteristics for a typical JFET is shown in Figure 5a.

The beginning of each curve of I_D (drain current) against V_{DS} (drain-source voltage) for a constant value of V_{GS} (gate voltage relative to the source) is linear. In other words, for each V_{GS} a different value of R_{DS}, the drain-source resistance, will be obtained for small signals.

Thus, operation in the small signal regions of the characteristics can provide controllable resistance.

If the audio amplifiers are provided with variable passive resistances at the input of each final stage,

the arrangement could be as shown in Figure 6. As shown, each channel has a series connection of two resistors, R_2,R_1 and R_1,R_2 respectively. The output voltage is taken from the interconnection node of each resistor pair in each case.

Increasing the balance in one side decreases the balance for the other side.

This sort of arrangement is often used for conventional manual stereophonic balance adjusting, and uses a logarithmic potentiometer, so that linear changes in the balance potentiometer will compensate for the logarithmic sensitivity of the ear. However, for automatic balance adjusting the changes will be linear.

In order to implement the variable resistances with JFETs, the JFETs all have to be compatible with the same characteristics and will preferably have zero threshold voltages. A possible arrangement is shown in Figure 7. In this case, the passive resistors of Figure 6 are replaced by respective JFETs Q_1, Q_2, Q_3 and Q_4 which are connected in pairs for respective channels and are supplied with current from respective preamplifiers of the amplifier block 350. When V_T is the threshold voltage of each JFET, and V_{gs1} and V_{gs2} are the signals applied to the gates of respective JFETS, it may be shown that:.

$$V_{out}/V_{in} = (V_{gs1} - V_T)/(V_{gs1} + V_{gs2} - 2V_T).$$

Since in our case $V_2/V_1 = d_2/d_1$, gate control signals V_{gs1} and V_{gs2} for the JFETs may be derived as shown in Figure 3. The outputs are taken from the central nodes of the respective JFET pairs and are connected to respective power amplifiers of block 350.

It would also be possible, on the basis of the timed intervals and the known velocity of sound in air (331.7 metres per second) to derive a measure of the distance travelled by the two ultrasonic signals. This distance is of course twice the distance between the loudspeaker unit 400 or 500 and the listener 600. From this distance value, a power compensation circuit could compute a measure of the sound intensity at the position of the listener 600 caused by the loudspeaker associated with the relevant unit 400 or 500. A similar measurement and computation process could be effected for each loudspeaker 100 and 200 and associated unit 400 and 500. From the resulting information, bearing in mind that received sound intensity falls off as the square of the distance, an appropriate adjustment could be made by a control device contained within the unit 300 to maintain the intensity from each speaker at a constant value as the distance changed. It will be noted that the position of the listener could also be determined from the distance information, since it is located at the intersection of two circular arcs centred at respective units 400 and 500 of radius equal to the computed distances. It is true that this information gives two possible positions for the listener, but one of these positions is excluded in the normal case since the loudspeakers are usually mounted adjacent a wall and one of the positions would be outside the room. However, it is not actually necessary to compute the coordinates of the listener in order to correctly adjust the stereo balance or the absolute volume.

Since there are two separate loudspeakers, it is clearly necessary to provide some means for

distinguishing the signals from unit 500 from those from unit 400. This may be achieved in one of at least two ways. In one possible method, unit 400 and unit 500 operate in differing frequency bands.

It will be appreciated that when units 400 and 500 are built into the housings of respective speakers 100 and 400 all necessary control and data lines may be conducted within a common speaker cable, preferably the same cable utilized for supplying the audio signals to the speaker.

Control of the system may be effected either by dedicated electronic control logic circuits as illustrated in conjunction with Figure 2, or by means of a microprocessor in conjunction with an appropriate program stored in a read-only memory.

Considering the implementation of the multi-phonic balancing system in more detail, there are three important factors to consider:

1. The source (the speaker).

2. The environment (closed space).

3. Symmetrical organization.

For simplicity, the description and analysis for the first two factors will refer to one speaker.

1. <u>The Source</u>

For an isotropic source (ideal source) transmitting in an isotropic medium (open space), the sound power

(P) at each point at distance D from the source will be proportional to $1/D^2$.

Figure 8 shows a radiation pattern of a non-omnidirectional (approx. 180°) speaker.

Although at any instant there will be a difference between the frequencies transmitted by respective speakers, the transmission loss at distance D from each speaker may be regarded as about the same for each speaker because the difference in frequency is usually slight and because there is a logarithmic relationship between the frequency and the transmission loss.

Therefore, the loss characteristic as a function of frequency need not be taken into account.

The speaker is a non-isotropic source because it has a non-zero size. This also need not be taken into account because of the following:

a) its size is negligible compared to the size of the space occupied; and

b) in stereophonic systems the supposition is made that all speakers have the same volume, size and structure.

2. <u>The Environment (Transmission Medium)</u>

When the speaker is transmitting inside a closed space, there are reflections from the floor, the walls, the ceiling and other objects, and the assumption that P is proportional to $1/D^2$ may not be exactly true. In order that this relation would be

applicable, reflections should be avoided as much as possible. For example, when the speaker is inside a room that is well carpeted, has curtains covering all the walls and has an acoustic ceiling, the reflection effect would be negligible. In this case it can be assumed that we are dealing with an isotropic medium, and so we will assume this for the user.

3. Symmetrical Organization

In a multiphonic (e.g. stereophonic or quadraphonic) system, it would be helpful to keep more or less symmetrical organization of objects in the room with reference to each speaker.

REFLECTIONS AS A RESULT OF THE ULTRASONIC WAVE

Reflections of the ultra-sound waves from surrounding objects of course have to be considered.

In order that the system can ignore reflections from other objects in the listening room, it will be necessary for the listener initially to identify himself as a target to the system. In other words, the listener must indicate his position before the balancing function can begin in order to initiate the distance measuring operation. Once the system has identified the location of the listener, and has thus "locked-on" to the listener's position, further detection of the listener's position may be achieved by identifying echos from locations where no echos were previously detected. This would indicate that a new object had moved to the location from which the echo was received. If this position is sufficiently near to the listener's original position and no echo is received from the listeners previous position, it

may be assumed that the listener has moved to the new position. The balance could accordingly be adjusted. Echos from other objects, especially those which are not moving, may be ignored by the system.

A further possibility not mentioned in the foregoing would be to provide one or more motors for adjusting the orientation of each loudspeaker. Thus, when the position of the listener has been calculated, the motors could be controlled in order to point the respective speakers towards the listener. This would ensure optimum listening for the user, bearing in mind the radiation pattern for a non-omnidirectional speaker illustrated in Figure 8.

DISTANCE AND SENSITIVITY FOR THE ULTRA-SOUND WAVE

The transmission loss for ultra-sound wave is given by the following relationship:

$$T_{L(dB)} = 20\log F + 20\log D + K,$$

where T_L is the transmission loss, F is the frequency, D is the distance travelled and K is a constant depending on the specific surface density of the environment.

From this relation, the required transmission power can be derived as follows:

$$T_L \leqslant (\text{Transmission power})_{dB} - (\text{Receiving sensitivity})_{dB}$$

Various types of ultrasonic transducers are commercially available, such as those available from

International Specialists Inc and referred to as the Pulse Transit (PT) type which includes a transmitter and a receiver and can operate over a frequency range of from 20kHz to 250kHz. Transducers with various case diameters, e.g. from 12mm to 24 mm, may be obtained.

Many modifications will occur to those skilled in the art and it is intended that all such modifications are included within the scope of the present invention as defined by the appended claims.

For example, distance measuring devices other than ultrasonic devices may be employed. One possibility would be infrared range-finding devices such as employed in auto-focus cameras.

CLAIMS

1. Apparatus for maintaining the subjective balance for a listener amongst the channels of a multi-phonic (e.g. stereophonic) sound reproduction facility as the listener moves around a space served by the facility, comprising: measuring devices for respective speakers arranged to determine the relative distances of the speakers from the listener by means of signals reflected from the listener; and control means arranged to vary the respective volumes of sound reproduction from said channels according to said relative distances.

2. Apparatus according to claim 1 wherein each measuring device is adapted to transmit an ultrasonic signal for reflection from the listener and to detect a resulting reflected signal.

3. Apparatus according to claim 2 wherein each said measuring device is arranged to transmit ultrasonic pulses and to detect ultrasonic echo pulses reflected from the listener.

4. Apparatus according to any one of claims 1 to 3 wherein each measuring device comprises: transmission means for transmitting pressure wave signals; receiving means for receiving reflected signals; and timing means coupled to said transmission means and to said receiving means and arranged to time the time interval between transmission of each signal by said transmission means and receipt of the corresponding reflected signal by said receiving means.

5. Apparatus according to claim 4 wherein each said transmission means is arranged to transmit an ultrasonic signal.

6. Apparatus according to claim 4 or 5 wherein said receiving means is positioned at the same location as said transmission means.

7. Apparatus according to any one of claims 4 to 6 wherein the measuring devices are arranged to transmit at mutually differing frequencies.

8. Apparatus according to any one of claims 4 to 6 wherein the measuring devices are arranged to transmit at mutually different times.

9. Apparatus according to any one of claims 4 to 8 wherein computing means is coupled to said timing means and is arranged to compute position data in respect of said listener from said time intervals.

10. Apparatus according to claim 9 wherein said computing means is arranged to further compute a control signal based on said position data, control means being provided for controlling the power supplied to said speakers in response to said control signal in a manner such as to equalize the audio intensities from respective loudspeakers at said listener location.

11. Apparatus according to any one of the preceding claims wherein said measuring devices are within respective housings of said loudspeakers.

12. A method for maintaining the subjective balance for a listener between the channels of a multi-phonic

(e.g. stereophonic) sound reproduction facility as the listener moves around a space served by the facility, in which method measuring devices for respective speakers determine the relative distance of the speakers from the listener by means of signals reflected from the listener, and the respective volumes of sound reproduction from said channels are varied accordingly.

13. Apparatus for maintaining the subject balance for a listener amongst the channels of a multi-phonic (e.g. stereophonic) sound reproduction facility substantially as hereinbefore described with reference to the accompanying drawings.

14. A method for maintaining the subject balance for a listener amongst the channels of a multi-phonic (e.g. stereophonic) sound reproduction facility substantially as hereinbefore described with reference to the accompanying drawings.

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